APPARATUS AND METHOD FOR ANALYZING AN ELECTRO-ACOUSTIC SYSTEM

TECHNICAL FIELD

The present invention relates generally to audio test equipment, and more particularly, to a system and method for analyzing the distance of the acoustic center of an acoustic transducer and its polarity.

BACKGROUND OF THE INVENTION

Acoustic transducers of a conventional electro-acoustic system, such as loudspeakers, play a fundamental and significant role in an audience's listening experience. The loudspeakers are, in fact, the critical link between the electrical signal representing audio information and the resulting audio signal heard by the listener. Consequently, the performance of an electro-acoustic system may be severely limited by its loudspeakers. The loudspeakers of an electro-acoustic system must reproduce sound throughout the audio spectrum, which is typically considered to be from 20 Hz to 20,000 Hz. It is difficult for a single loudspeaker to accurately reproduce sound over the entire frequency range. Several loudspeakers are typically used in such a situation to provide adequate volume and coverage of the audio spectrum. Each loudspeaker is dedicated to reproducing sound for a particular frequency range, and the complement of loudspeakers are coordinated through a crossover network. However, electro-acoustic systems having multiple loudspeakers pose a particular challenge to the audio engineer.

In the case where the multiple loudspeakers are physically disassociated and may be placed at different positions throughout the listening area, the issue of time coherency between the audio signal generated by each loudspeaker becomes particularly significant. That is, the position of each loudspeaker relative to one another is related to the difference in time for the resulting audio signal produced by the respective loudspeaker to reach a particular point in the listening area. It is quite often the case where the poor quality of sound is not due to the quality of the loudspeaker itself, but is a result of the multiple loudspeakers not being aligned in time effectively for the

majority of the audience. Thus, one of the goals of an acoustic engineer is to arrange and correct for the various positions of the loudspeakers of the electro-acoustic system in order to produce a coherent wavefront. The benefits of time correcting multiple loudspeakers located in acoustic space include minimized comb filtering, reduction of reverberant field, and increased intelligibility of the acoustic signal.

Practically speaking, it is often not possible to physically position multiple loudspeakers in relation to one another to produce a coherent wavefront. The positioning of loudspeakers may be limited by the physical space available for placement of the loudspeakers, as well as the size and shape of the listening area in which the loudspeakers are placed. To accommodate the various placement of loudspeakers in relation to one another, programmable electronic delay circuits have been used to correct for time disparities between the loudspeakers. The delay circuits may be programmed to delay the arrival of the stimulus signal to one loudspeaker, with respect to another loudspeaker, so that the difference in their relative position may be compensated by the programmed time delay. Thus, a more coherent wavefront of the resulting audio signal may be produced. However, in order for this method of time correction to produce sufficient results, it is necessary to determine the relative position of the multiple loudspeakers. The relative position of the multiple loudspeakers may be determined by the relative time delay of the acoustic signals of each loudspeaker.

One method of determining the relative time delay of the multiple loudspeakers is to physically measure the distance from each of the loudspeakers to a point located in the listening area. The relative time delay of each loudspeaker may be calculated from the resulting measurement, and used to program the appropriate delay times of the delay circuits. However, this method does not acknowledge the fact that the distance of each loudspeaker should be measured from its respective "acoustic center." The acoustic center of a loudspeaker is a term used to note the actual sonic origin of sound from the loudspeaker. The acoustic center is typically located further away than the loudspeaker itself. Thus, measuring the physical distance of the actual loudspeaker will not necessarily coincide with the physical distance of its acoustic

center. In programming the delay times to produce a coherent wavefront, the distance should be measured from the acoustic center.

Further complicating the measurement is the fact that the acoustic center of a loudspeaker is frequency dependent. Due to the electro-mechanical nature of conventional loudspeakers, its acoustic center shifts depending on the frequency of the audio signal being produced. Consequently, the relative distances of multiple loudspeakers will change throughout the audio spectrum. Another factor that should be considered, but cannot be determined from physical measurement, is additional delay introduced by the electro-acoustic system itself, for example, digital signal processing of the stimulus signal prior to providing the resulting analog signal to the loudspeakers.

Another factor affecting the quality of sound the audience experiences is the polarity of a particular loudspeaker with respect to the stimulus signal, as well as to the other loudspeakers. The polarity of a loudspeaker is determined by its connection to the power amplifier of the electro-acoustic system. Two loudspeakers connected to have opposite polarities will produce audio signals 180 degrees out of phase. Consequently, the resulting audio signals may destructively interfere with one another and affect the overall sound quality. Determining the polarity of a particular loudspeaker by visually inspecting its connection to the power amplifier may not be practical if the loudspeaker is located in a position that is difficult to reach. For example, the speaker may be located high above the listening area, or may be mounted into a wall. In either case, visually inspecting the connection of the loudspeaker will not be easy.

There currently exists analysis equipment for evaluating various performance characteristics of an electro-acoustic system and its loudspeakers. One such system is described in United States Patent No. 5,555,311 to Reams, issued September 10, 1996. The system described in the Reams patent can determine, among other things, the bandwidth, the thermal limit, and the group delay of an electro-acoustic system. A Tef System and SIM System II are some additional examples of analysis equipment. These tools perform Fast Fourier Transforms on an impulse stimulus yielding a complete time and frequency analysis. Another measurement tool for

loudspeaker evaluation and room acoustic which uses maximum length sequences is known as MLSSA. However, determining the time delay of the electro-acoustic system and its loudspeakers using these equipment involve interpreting data provided in a format more suited for discerning other measurement data. For example, measurement data may be provided in the form of a graphical representation of an impulse response. Determining the time delay involves interpreting the graphical information, which may require special training to understand the resulting measurement data. Furthermore, the existing analysis equipment often involve complicated and time-consuming setup procedures, which may also be carried out only by specially trained technicians.

SUMMARY OF THE INVENTION

The present invention is directed to a method and apparatus for measuring the time-of-flight of an audio signal generated in response to a stimulus signal by an electro-acoustic transducer, such as an audio loudspeaker, to a point of measurement by correlating the audio signal to a delayed signal having similar characteristics as the stimulus signal. The measurement identifies the total overall delay from the time the stimulus signal is generated to the time the resulting audio signal is detected. Thus, any system delays, such as those due to signal processing, have already been accounted for by the measurement. Some or all of the measurement process may be automated through software programming. The distance of the acoustic center of the transducer can be calculated from the measured time-of-flight. The resulting comparison between the audio signal and the delayed pseudo random noise signal may also be used to determine the polarity of the transducer with respect to the stimulus signal.

BRIEF DESCRIPTION OF THE DRAWINGS

Figure 1 is a block diagram of an analysis system according to an embodiment of the present invention.

Figure 2A-B is a flowchart showing an operation of the analysis system of Figure 1 for calculating the distance of the acoustic center of a electro-acoustic transducer according to an embodiment of the present invention.

Figure 3A-B is a flowchart showing an operation that may be substituted into the operation shown in Figure 2A-B for determining the peak correlation according to an embodiment of the present invention.

Figure 4A-B is a flowchart showing an operation that may be executed in addition to the operation shown in Figure 2A-B for more precisely determining the peak correlation according to an embodiment of the present invention.

DETAILED DESCRIPTION OF THE INVENTION

Shown in Figure 1 is an analysis system 10 according to an embodiment of the present invention. The analysis system 10 is coupled to a conventional electro-acoustic system 2, having a power amplifier 4 and an electro-acoustic transducer 6, such as an audio loudspeaker. The analysis system 10 measures the delay time for a stimulus signal applied to the electro-acoustic system 2 to be detected at a measurement point. The delay time is in turn used by the analysis system 10 to calculate the distance between the transducer 6 and the measurement point for a particular measurement frequency. The analysis system 10 may be programmed to automatically perform the measurement, or the measurement may be manually performed by an operator.

Various signals may be used for the stimulus signal. Generally, the stimulus signal should have unique characteristics so that a delay time may be determined from correlating the audio signal generated in response to the stimulus signal with a delayed version of the stimulus signal. A signal such as a simple sine wave having a constant magnitude and period would not be appropriate. However, music signals, noise signals, and the like could be used as the stimulus signal of the analysis system 10. In the case where music is used as the stimulus signal, the analysis system 10 may perform the measurement in real-time. Accordingly, the present invention is not limited to the use of a particular stimulus signal.

A preferred embodiment of the analysis system 10 uses pseudo-random pink noise as the stimulus signal. A signal is characterized as being pseudo-random when the random noise sequence can be repeated, provided the same seed value is used to begin the sequence. That is, the pseudo-random noise signal has a distinct, but repeatable pattern. As will be explained in greater detail below, a pseudo-random signal facilitates the use of two separate signal generators to provide the stimulus and delayed stimulus signals. A pseudo-random noise signal can be devised to cover all or any part of the audio spectrum. Thus, the frequency range of the noise signals may encompass the bandwidth of the electro-acoustic system 2. The length of the pseudo-random sequence will be greater than the measurement time, and will consequently appear infinitely long to the electro-acoustic system 2.

The operation of the analysis system 10 is controlled by a conventional microprocessor 12. The microprocessor 12 provides activation signals through AND gates 13a and 13b to pseudo-random pink noise generators 14 and 16, respectively, to initiate a pseudo-random noise sequence. A clock circuit 18 provides through a buffer circuit 19 a clock signal that is used as the clock rate for the noise generators 14 and 16. The pseudo-random noise signal generated by the noise generator 14, NOISE1, is provided to a test signal connector 20. The electro-acoustic system 2 is connected to the test signal connector 20 to receive the NOISE1 signal as a stimulus signal. The audio signal generated by the transducer 6 in response to the NOISE1 signal is detected by a conventional microphone 22 coupled to the analysis system 10 through a connector 23. The resulting electrical signal output by the microphone 22 is applied to a band-pass filter 26 through a conventional preamplifier 24. Although the electro-acoustic system 2 is shown in Figure 1 as having only one transducer 6, several transducers 6 may be included in such an electro-acoustic system 2. However, additional transducers 6 have been omitted from Figure 1 in order to simplify explanation of the analysis system 10.

The noise generator 16 generates a noise signal, NOISE2, having the same pseudo-random sequence as the NOISE1 signal by using the same seed value to begin the NOISE2 sequence. As will be explained in greater detail below, the NOISE2 signal is initiated at a time after the NOISE1 signal is initiated. As explained below, the

delay time of the NOISE2 signal with respect to the NOISE1 signal will be used by the analysis system 10 to determine the physical distance between the acoustic center of the transducer 6, at a particular frequency, and the microphone 22. The length of the delay time may be automatically adjusted by the analysis system 10, or manually adjusted by the operator making the measurement.

The NOISE2 signal is provided to a band-pass filter 30 having similar filter characteristics as the band-pass filter 26. The filter characteristics of filters 26 and 30 should be similar in order to maintain the phase and time relationship between the audio signal detected by the microphone 22 and the NOISE2 signal. Additionally, both the band-pass filters 26 and 30 have similar low-quality factors ("low-Q") in order to weight the frequencies near the band-pass frequency of the filters 26 and 30, while retaining the essential randomness of the signals being filtered. The band-pass frequency of both filters 26 and 30 are controlled by the frequency of a signal generated by a filter oscillator 36. The operating frequency of the oscillator 36 is controlled by the microprocessor 12 by data provided over a data bus 37. As will be explained below, the operating frequency, and consequently the band-pass frequency of filters 26 and 30, may be selected in order to facilitate measuring, at a particular frequency, the distance of a transducer of the electro-acoustic system 2, which may use multiple transducers to provide full coverage of the audio spectrum.

The output of filters 26 and 30 are provided to a comparison circuit 38. The comparison circuit 38 includes a conventional mixer 40, such as a four quadrant analog multiplier, coupled to receive the filtered signals of the band-pass filters 26 and 30. The mixer 40 compares the two filtered signals and generates a correlation signal having a magnitude indicative of the real time signed product of the two signals provided by the filters 26 and 30. The output of the mixer 40 is applied to a low-pass filter 42 in order to average the signed product output signals that occur wave by wave at audio frequencies in real time. The comparison circuit 38 further includes an analog-to-digital converter ("ADC") 46, that is coupled through a signal buffer 44 to the output of the low-pass filter 42. The ADC 46 is used to sample the output signal and provide digital data representing the value of the output signal to the microprocessor 12 on the

data bus 37. As will be explained in greater detail below, the output of the low-pass filter 42 is sampled and averaged over a period of time to produce a signed average value indicative of the correlation between the audio signal produced by the transducer 6 and the NOISE2 signal. Generally, the signed average value of the mixer 40 is positive if the two signals are in time and have the same polarity, negative if the two signals are in time and have opposite polarities, and nearly zero if the two signals do not correlate.

A set of signed average values are generated for a range of delay times, and are stored with the associated delay times in a memory 50. The microprocessor 12 evaluates the stored data and determines the signed average value corresponding to the peak correlation between the audio signal produced by the transducer 6 and the NOISE2 signal. The delay time associated with the selected signed average value is used by the microprocessor 12 to calculate the physical distance between the acoustic center of the transducer 6 at the particular frequency and the microphone 22. The resulting distance value is provided by the microprocessor 12 to a conventional display driver 52 for displaying the results on a display 54.

The analysis system 10 also includes a number of other components and signal lines that have been omitted from Figure 1 in the interests of brevity. For example, a digital-to-analog converter ("DAC") coupled to the microprocessor 12 provides the analog signal through a multiplexer (not shown) to the oscillator 36 to establish its operating frequency. The multiplexer is under the control of the microprocessor 12, and facilitates the use of the DAC for other control signals. An analysis system similar to the analysis system 10 is described in greater detail in the aforementioned Reams patent, which is incorporated herein by this reference.

Figure 2 illustrates the process by which the analysis system 10 can be employed to determine the delay time of a stimulus signal through the electro-acoustic system 2. As a matter of convenience in explaining the process illustrated in Figure 2, the stimulus signal employed by the analysis system 10 is a pseudo-random pink noise signal. However, as mentioned previously, other signals, such as music, may be used as the stimulus signal as well. Based on the delay time determined by the analysis system

10, the physical distance from the acoustic center of the transducer 6 and a point of measurement may be calculated for a particular frequency. Some or all of the steps described herein may be automatically performed through software programming of the analysis system 10. Once the measurement equipment is in place, and the analysis system 10 is connected to the electro-acoustic system 2, such software programming facilitates automatic calculation of the physical distance of the acoustic center of the transducer 6. However, measurement using manual adjustments by the operator may be made available where such feature is desired.

At a start 100, shown in Figure 3, the analysis system 10 has been connected to the electro-acoustic system 2 by connecting the power amplifier 4 to the test signal connector 20 to receive the NOISE1 signal, and the microphone 22 connected to the connector 23 has been positioned at the point of measurement in the listening area. In step 102, a particular measurement frequency, f_M, is selected at which the delay time is determined. As a mentioned previously, the phase response of the conventional transducer 6 is dependent on frequency, and the acoustic center of the transducer 6 varies with the frequency of the signal being reproduced. Therefore, a particular measurement frequency should be selected in order to obtain relevant measurement data.

As mentioned previously, the measurement frequency should be selected so that when multiple transducers 6 are being used to provide full coverage of the audio spectrum, the delay time of a stimulus signal reproduced by a particular transducer may be measured. A suggested frequency at which to measure a particular electro-mechanical transducer is at its "energy center frequency," f_{EC}. This is defined as the frequency at which the transducer produces equal acoustic energy above and below the frequency over the 20 Hz-20 kHz frequency spectrum. Coincidentally, the particular transducer reproduces a signal having minimal phase error with respect to the input signal at the f_{EC}. However, while measuring at the f_{EC} of a transducer is preferred, measurements may be made at other frequencies. For example, in an electro-acoustic system having separate loudspeakers for low, mid, and high frequencies, the desired f_M may be selected at the crossover frequencies of the separate loudspeakers, or in the

frequency region where the frequency overlaps. Choosing such an f_M permits measurement of the overall delay, including phase shift introduced by the crossover. The resulting measured time interval can then be used to accurately align the resulting audio signal in the portion of the frequency spectrum the separate loudspeakers are required to reproduce, namely, in the crossover region.

After the f_M has been selected, in step 104 the microprocessor 12 initiates the noise generator 14 to begin generating a pseudo-random pink noise signal, NOISE1. As discussed previously, the pseudo-random nature of the NOISE1 signal is characterized by a random sequence that may be reproduced, provided that an identical seed value is used to initiate the signal. The NOISE1 signal is provided to the input of the power amplifier 4 of the electro-acoustic system 2 as a stimulus signal to drive the transducer 6 in step 106. In step 108, the resulting audio signal produced by the transducer 6 is detected by the microphone 22 positioned at the point of measurement, and pre-amplified by the pre-amplifier 24 of the analysis system 10. Subsequently, in step 110, the pre-amplified signal is filtered through the low-Q band pass filter 26 having a band pass frequency, f_C, that is equal to the f_M. As mentioned previously, the low-Q nature of the bandpass filter provides for the weighting of the frequencies near the f_M while still retaining the essential randomness of the noise signal.

From step 104, the analysis system 10 performs steps 112 and 114, which relate to a second pseudo-random noise signal generated by the noise generator 16, NOISE2. In step 112, generation of the NOISE2 signal is delayed with respect to the NOISE1 signal. The delay time selected should take into consideration the approximate time-of-flight of a signal produced by the transducer 6 and detected by the microphone 22 positioned at the point of measurement. The time-of-flight of an audio signal produced by the transducer 6 to the microphone 22 positioned at the point of measurement may be approximated by providing a sine ping as a stimulus signal to the electro-acoustic system 2. The analysis system 10 measures the time delay between generating the sine ping and receiving the resulting audio signal produced by the transducer 6 to determine the approximate time-of-flight. Such a technique can locate the acoustic center of the transducer 6 within a few wavelengths. This is sufficient for

determining an appropriate initial delay time for the NOISE2 signal, however, the measurement lacks the desired accuracy necessary to produce a coherent wavefront in a multi-transducer system. Furthermore, the polarity of the transducer 6 cannot be determined using the aforementioned technique.

As will be explained in more detail below, a more accurate measurement of the distance will be determined by the analysis system 10, by correlating the audio signal produced by the transducer 6 and the NOISE2 signal over a range of delay times which includes the approximate time-of-flight. As a practical matter, in order to facilitate measurement, a suggested delay time at which to begin generating the NOISE2 signal is the sum of the approximate time-of-flight and 1.5 waveperiods at the f_M. That is,

$$\text{delay time of noise signal 2} = (\text{approximate time-of-flight} + 1.5T_{fM})$$

Measurement of the approximate time-of-flight and selection of an initial delay time may be performed by the analysis system 10 via software programming in order to automate the delay time measurement.

A person of ordinary skill in the art will appreciate that generating the NOISE1 and NOISE2 signals may be implemented in a variety of ways. As illustrated in Figure 1, the two separate pseudo-random pink noise generators 14 and 16 used to generate the NOISE1 and NOISE2 signals, respectively, are coupled to receive activation signals from the microprocessor 12. The activation signal initiating the noise generator 16 is delayed by the microprocessor 12 with respect to the activation signal initiating the noise generator 14. Thus, the delay time of the NOISE2 with respect to the NOISE1 signal is a result of the microprocessor 12 activating the second noise generator 16 at a time later than activating the first noise generator 14. However, the NOISE1 signal and the NOISE2 signal may also be produced using a single pseudo-random pink noise generator and a variable delay circuit (not shown) having a delay time automatically adjusted by the microprocessor 12, and additionally, or alternatively, manually adjusted by the operator. The output of the single noise generator is provided to the power amplifier 4 of the electro-acoustic system 2, as well as to the input of the variable delay circuit. The output of the variable delay circuit provides the NOISE2

signal. In this case, the delay time of the NOISE2 signal is a result of adding a delay time controlled by the microprocessor 12 to a common noise signal. The use of a single noise source and a microprocessor controlled variable delay circuit facilitates using music, live or recorded, as a stimulus signal. These examples are provided for the purposes of illustration, and, because various methods of producing a delayed noise signal with respect to another noise signal are well known in the art, are not meant to limit the scope of the present invention.

Subsequently, in step 114, the NOISE2 2 is filtered through the low-Q band pass filter 30 having its f_C equal to the f_M. As mentioned previously, the band pass filter 30 should have filter characteristics very similar to those of the band pass filter 26 used for the pre-amplified signal in order for the NOISE2 signal to have the same weighted frequency relationship as the pre-amplified signal.

The pre-amplified signal detected by the microphone 22 and the NOISE2 signal are compared in step 120 by providing each signal to the conventional mixer 40, such as a four-quadrant multiplier. The mixer 40 produces an output signal having a polarity and voltage value indicative of the product of the pre-amplified signal and the NOISE2 signal. As mentioned previously, the multiplier 40 will produce a positive output signal if the two signals being compared are in time and have the same polarity, and a negative output signal if the two signals are in time but have opposite polarities. Consequently, if the voltage value of output signal of the multiplier is sampled and converted from an analog to digital value over a predetermined length of time, the resulting signed average value will be indicative of the correlation between the two signals input to the mixer 40. If the two signals do not correlate, the output of the multiplier will average to nearly zero over the sample period. The resulting analog output of the mixer 40 is sampled and converting to digital values by the ADC 46. Analog-to-digital conversion is well known in the art, and will not be discussed in detail herein in the interest of brevity.

A person of ordinary skill in the art will appreciate that the output signal of the mixer 40 should be filtered by the low pass filter 42 to attenuate high frequency components of the output signal prior to sampling. Furthermore, sampling should not

begin until there has been sufficient time for the output of the low-pass filter 42 to settle. Such details are within the knowledge of those ordinarily skilled in the art, and may be resolved without undue experimentation. Thus, discussion of such details have been omitted from herein.

As mentioned previously, a signed average value is generated by sampling the magnitude of the output signal of the mixer 40, and averaging the sampled values. Although the specific number of samples that should be made is not limited to a specific number, it has been determined that making 128 samples at 20 ms intervals provides sufficient data to produce a signed average value that is indicative of the correlation of the pre-amplified signal and the NOISE2 signal. However, as a person of ordinary skill in the art will appreciate, the number of samples taken, and the samples rate may be increased or decreased, depending on the accuracy and measurement time desired, and the stimulus signal used by the analysis system 10. That is, other types of stimulus signals may need more samples to be taken in order to obtain accurate results.

As will be explained in greater detail below, a resulting signed average value is determined for a series of delay times which includes the approximate time-of-flight. That is, for each delay time there will be an associated signed average value that is indicative of the level of correlation at that delay time. From the accumulated data, the analysis system 10 will determine the signed average value corresponding to the peak correlation between the pre-amplified signal and the NOISE2 signal. The delay time associated with that signed average value is used to calculate the distance from the acoustic center of the transducer 6 to the microphone 22. Where the delay time is to be manually adjusted, the signed average value corresponding to the peak correlation may be obtained through a method of manual metering. For example, the operator can adjust the delay time until a peak value is displayed on a meter (not shown), indicating that the delay time of peak correlation has been obtained. Providing a manual metering adjustment for the delay time is well known in the art, and will not be discussed in detail herein.

Following the comparison of the pre-amplified signal and the NOISE2 signal of step 120, the analysis system 10 records the signed average value and its

associated delay time in the memory 50 at step 122. The analysis system 10 begins the process of recording signed average values for a series of delay times in steps 124 and 126. At step 124, the determination is made whether a predetermined number of signed average values and associated delay times have been recorded. If the number has not been reached, the delay time of the NOISE2 signal is incrementally stepped at step 126 so that at steps 120 and 122, a new signed average value can be generated and recorded for the new delay time. The direction in which the delay time is stepped will be determined by the initial delay time of the NOISE2. That is, in the case where the initial delay is greater than the approximate time-of-flight, the delay time should be stepped downward to incrementally reduce the delay time with each successive step. On the other hand, if the initial delay time of the noise signal 2 is less than the approximate time-of-flight, the delay time should be incrementally increased with each successive step. In the present example, where the initial delay time of the NOISE2 signal is the sum of the approximate time-of-flight and 1.5 waveperiods at the f_M, the delay time will be incrementally reduced. The process of incrementally stepping the delay time, and generating and recording a signed average value for each new delay time, repeats until the number of signed average values reaches the predetermined number.

Determining the predetermined number of signed average values that the analysis system will generate and record, as well as determining the increment by which the delay time is stepped, depend on a variety of factors. One factor that should be considered is the f_M. Lower frequencies, which have longer wavelengths, require larger increments in order for the distance measurement to be accurate, and for the measurement to finish within a reasonable time frame. Consequently, the analysis system 10 is preferably programmed with incremental delay times that are related to the f_M. The number of signed average values that are generated should take into consideration the number that will provide an accurate measurement without taking an unduly long time. As an example, the analysis system 10 may be programmed to step the delay time of the NOISE2 signal in increments of $1/24T_{fM}$, for 96 successive delay steps. As a result, the signed average value will be generated and recorded for time

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span of $4T_{fM}$. It has been found that this time span is sufficient to produce accurate results while finishing within a reasonable time frame.

As a person of ordinary skill in the art will appreciate, the specific increments of delay times, whether the successive incremental steps decrease or increase the delay time, or the number of incremental delay steps taken, are details that may be modified, but nevertheless remain within the scope of the present invention.

After a sufficient number of signed average values have been generated and recorded through steps 120-126, the analysis system 10, in step 130, will evaluate the recorded signed average values for the one corresponding to the peak correlation between the pre-amplified signal and the NOISE2 signal. The analysis system 10 begins with the signed average value associated with the lowest delay time and proceeds to each successive increasing delay time. As mentioned previously, the signed average value represents the magnitude of correlation between the two signals. Consequently, the delay time associated with the signed average value determined to be the peak correlation is the time for a stimulus signal provided to the electro-acoustic system 2 to be reproduced by the transducer 6 and detected at the point of measurement. In step 132, the physical distance between these two points is calculated by the analysis system 10 based on this delay time.

In addition to calculating the physical distance from the acoustic center of the transducer 6 and the point of measurement, in step 134, the polarity of the transducer 6, with respect to the stimulus signal, may be determined by the sign of the signed average value that was determined to be the peak correlation. A positive average value indicates that the transducer 6 is connected to the power amplifier 4 to generate audio signals having the same polarity as the stimulus signal. Conversely, a negative average value indicates that the polarity is opposite of the stimulus signal.

In step 136, the resulting distance value and polarity of the transducer 6 are provided by the microprocessor 12 to a conventional display driver 52 for displaying the results on a display 54. The process of measuring the distance from the acoustic center of a transducer at a particular frequency to a point of measurement ends at step 140.

A method that may be used to perform step 130 of Figure 2 to determine which of the recorded signed average values corresponds to the peak correlation is described in detail in the flowchart shown in Figure 3. At a start 160, a sufficient number of signed average values and associated delay times have been recorded. In evaluating the recorded signed average values, the analysis system 10, in step 162, identifies a first peak average value exceeding a threshold value. The threshold value should be high enough to prevent the analysis system 10 from selecting a minor peak value as the first peak average value, while low enough to ensure that a peak average value will be detected. The value of the threshold voltage will be determined by the desired sensitivity of the analysis system 10.

In step 164, the analysis system 10 further identifies a second peak average value exceeding the threshold value, but having an opposite sign of the first peak average value. That is, a peak average value having the opposite polarity of the first peak average value. The magnitude of the first and second peak average values will be used by the analysis system 10 to determine the peak average value corresponding to the peak correlation of the audio signal produced by the transducer 6 and the NOISE2 signal.

At step 166, the determination is made whether the f_M is greater than or equal to 800 Hz. It has been determined through experimentation that the following algorithm can be used to accurately determine the signed average value corresponding to the peak correlation. In the case where the f_M is greater than or equal to 800 Hz, another determination is made at step 170 regarding whether the ratio of the first and second peak average values is greater than or equal to 1.8. If so, then the analysis system 10 proceeds to step 172, and determines that the larger of the two peak values corresponds to the peak correlation. Consequently, the associated delay time will be used in step 132 (Figure 2) to calculate the distance from the acoustic center of the transducer 6 to the point of measurement. However, if the resulting ratio is not greater than or equal to 1.8, the analysis system 10 proceeds to step 174, and determines that the peak average value having the lower associated delay time corresponds to the peak

correlation. The associated delay time will be subsequently used in step 132 to calculate the distance of the transducer 6.

In the case where the f_M is not greater than or equal to 800 Hz, the analysis system 10 determines at step 176 whether the ratio of the first and second peak average values is greater than or equal to 3.0. As with step 170, if the condition is determined to be true in step 176, the larger of the two peak average values will be selected in step 178 as corresponding to peak correlation, and the associated delay time will be used in step 132 to calculate the distance of the transducer 6. However, if the condition is false, then the peak average value having the lower associated delay time is selected in step 174 to calculate the physical distance instead. The analysis system 10 returns to the step of 132 (Figure 2) subsequent to finishing at step 180.

As mentioned previously, the particular values used in the algorithm of Figure 3 are provided for the purposes of illustration. A person of ordinary skill in the art will realize, however, that a method using different ratios could also be used.

Illustrated in Figure 4 is a flowchart describing in detail an additional method that may be incorporated into the flow chart of Figure 2 for more precisely determining the signed average value corresponding to the peak correlation of the pre-amplified signal and the noise signal 2. The steps shown in Figure 4 can be inserted between steps 130 and 132 of Figure 2. At a start 200, the analysis system 10 has already evaluated and determined the signed average value corresponding to the peak correlation. A starting point for the precision measurement is selected in step 202. In the present example, subsequent signed average values will be generated for increasing delay times. Thus, a suggested starting point is at the delay value determined in step 130 (Figure 2), less $1/6T_{fM}$. However, it will be appreciated that the particular starting point may be increased or decreased from the suggested value. In step 204, a signed average value at the starting point is generated. The signed average value is recorded as the initial recorded value at step 208 if it is determined at step 206 that it is the first signed average value that has been generated. Subsequently generated signed average values will be compared to the initial recorded value.

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Generating a signed average value at step 204 is performed in a manner similar to step 120 of Figure 2. That is, the filtered analog output of the mixer 40 is sampled by the ADC 46 and resulting digital values are averaged. The output of the mixer 40 was suggested to be sampled 128 times at 20 ms intervals. In comparison to step 120, the number of samples made may be increased and the interval lengthened in step 204 to provide a more accurate signed average value. For example, the output of the mixer 40 may be sampled 256 times at 40 ms intervals. The particular number of samples made, and sampling frequency, will be determined based on factors such as the desired accuracy of the signed average value, the length of test time desired, and the type of stimulus signals used by the analysis system 10. Consequently, other values for the number of samples taken and the sampling rate may be used.

The delay time of the NOISE2 signal is incrementally stepped in step 210, and another signed average value is generated. As an example, the analysis system 10 may be programmed to increase the delay time of the NOISE2 signal in increments of $1/36T_{fM}$. Since the initial recorded value was already recorded in step 208, the determination of step 216 compares the magnitude of the most recent signed average value with the magnitude of the initial recorded value. If the magnitude of the most recent average value is greater than the initial recorded value, then it will replace the previous initial recorded value in step 218, and will be subsequently used as the new initial recorded value. As illustrated in Figure 4, the analysis system will continue to incrementally step the delay time of the NOISE2 signal and generate a new signed average value through steps 204-210, until the maximum value of the average value corresponding to the peak correlation between the pre-amplified signal and the NOISE2 signal has been determined.

When the magnitude of the most recent average value is finally found to be less than or equal to the initial recorded value of step 218, a determination at step 220 is made as to whether the two most recent average values generated have been both less than or equal to the initial recorded value. If there has been only one occurrence of the average value being less than the initial recorded value, the delay time is again incremented and another signed average value is generated. However, if the two most

recent numbers have both been less than or equal to the initial recorded value, the two numbers are then compared to each other at step 224. If the two numbers, with respect to each other, continue to exhibit a diminishing trend, the determination of step 224 is true, and the first of the two most recent average values generated is selected as the signed average value corresponding to peak correlation between the pre-amplified signal and the NOISE2 signal.

The determinations made at steps 220 and 224 are a means of ensuring that the first average value determined to be less than or equal to the initial recorded value is not simply a diminishing fluctuation, but is truly indicative of the first measurement point at, or just slightly past, the peak signed average value corresponding to peak correlation. The analysis system 10 returns to step 132 (Figure 2) and completes the measurement at previously described.

From the foregoing it will be appreciated that, although specific embodiments of the invention have been described herein for purposes of illustration, various modifications may be made without deviating from the spirit and scope of the invention. For example, and as mentioned previously, the particular values provided herein have been by way of example. However, the particular sampling rates, number of samples taken, and starting points, among other things, may be selected according to the desired functionality of the analysis system 10, and do not necessarily need to be identically reproduced as described herein. Accordingly, the invention is not limited except as by the appended claims.